Cross Layer based Hybrid fuzzy ad-hoc rate based Congestion Control (CLHCC) approach for VoMAN to improve quality of VoIP flows

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Abstract **- Mobile environment pretense a number of novel theoretical and optimization issues such as position, operation and following in that a lot of requests rely on them for desirable information. The precedent works are sprinkled across the entire network layer: from the medium of physical to link layer to routing and then application layer. In this invention, we present outline solutions in Medium Access Control (MAC), data distribution, coverage resolve issues under mobile ad-hoc network environment based on congestion control technique using Transmission Control Protocol (TCP). In mobile ad-hoc network issues can arise such as link disconnections, channel contention and recurrent path loss. To resolve this issue, we propose a Cross Layer based Hybrid fuzzy ad-hoc rate based Congestion Control (CLHCC) approach to maximize network performance. Based on the destination report it regulates the speed of data flow to control data loss by monitoring the present network status and transmits this report to the source as advice. The source adjusts the sending flow rate as per the advice. This is monitored by channel usage, ultimate delay, short term throughput.**

Index Terms— MAC, MPSD, RTT, SIFS TCP,

I. INTRODUCTION

Mobile Ad-hoc Network (MANET) is a transient dynamic network formed by set of mobile nodes. Routing in MANTE is a complex and challenging task because of its dynamic nature, link stability and infrastructure less concept. A packet might traverse many intermediate nodes supporting dynamic link to reach the desired destination. Routing algorithm should generate feasible route by collecting routing status and route packets over the optimal route to support Quality of Service (QoS).

Voice over MANET is becoming important day by day as users demand to use real time applications. As Quality of Service (QoS) issues in infrastructure based networks still remain unsolved, it is a challenging task in MANET that needs to be solved for real time applications [12]. Increase in real time traffic increases network congestion. Congestion control policies are classified into three types namely window based, rate based and hybrid. Window based congestion control policy adjusts the congestion window as per the changing network status. Rate based congestion control policy increases or decreases the data rate of the sender as per the current status of the network [8]. Hybrid approach combines both the above discussed policies to control congestion.

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This paper is organized as follows: Section II describes the literature survey and Section III presents the methodology of Fuzzy based Congestion Control approach using voice over TCP. Section IV presents the result analysis of the new approach. Section V summarizes the most important simulation results and their interpretation. Finally, Section VI concludes this paper.

II. LITERATURE SURVEY

Security in mobile ad hoc network is hard to accomplish due to vibrantly changing and fully decentralized topology as well as the vulnerabilities and limitations of wireless data transmissions[15].A lot of research has since focused on mechanisms to improve TCP performance in cellular wireless systems[1]. In MANET, due to dynamic nature of the network and variable number hops between source and destination, the fairness and efficiency get degraded. It is also seen that the existing protocols are not good enough in setting its parameters like congestion window, congestion window limit, round trip time and the retransmission timeout timers [11]. The routing overhead of such an algorithm increases with the square of the number of mobile nodes in a MANET[2]. A hybrid routing algorithm that combines the merits of existing protocols that can be used to address this issue of growth in network size and load balancing whose behavior can be modified according to the size of network [9]. The effect of high bit error rate and route re-computation on the performance of TCP in mobile ad hoc network is analyzed [4]. In particular, it allows the routing protocol to operate more efficiently by reducing the control traffic in the network and simplifying the data routing [7]. In order to avoid congestion in the network, it is required to use an efficient congestion control algorithm for successful transmission of data throughout the network [13]. Numerous examinations have been done to enhance the execution of TCP at network layer by enhancing the routing strategy. M-ADTCP is another method that presents a Modified AD-hoc Transmission Control Protocol where the receiver detects the probable current network status and transmits that information to the sender as feedback. The sender behaviour is altered appropriately [6]. In this way, there is an extension to enhance the execution of TCP at transport layer [14]. TCP supplies end-to-end reliable delivery of data between an application process on one computer and the process running on another computer, by adding services on top of IP [5]. In order to use these limited resources efficiently, an intelligent routing strategy is required which should also be adaptable to the changing conditions of the network, like, size of the network, traffic density and network partitioning[10].

III. A.CLHCC METHODOLOGY

Application Layer VoIP based Communication
Transport Layer Congestion detection and Controller System
Network Layer Computation of Middle Packet Setback Distinction, Short Term Throughput, Delay
Physical and MAC Layer Computation of Channel business ratio

Figure 1. Cross Layer based VoMAN Architecture

We consider a mobile ad-hoc network with n nodes S (s1; s2; . . .sn;). These n nodes are randomly disseminated in a field. At random time periods, each node

can send event values (E_V) from the field at its location and sends them E_V to the destination. Nodes send a network wide broadcast message at every particular interval, to all nearby nodes in the network. Nodes built with a global positioning system (GPS) recipient at an exactly known position which is used to receive collected information from nearby nodes. The nodes are installed randomly and they may organize by them self as hop based jointly to complete a transmission sensed the environment. Detection of nearby nodes is accomplished by a periodic HELLO messages sharing among the nodes, produced at fixed interval. When a node receives HELLO message it updates the neighbor table NT.

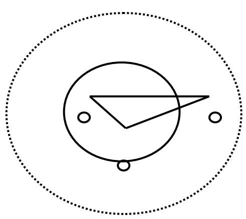

Figure 2. Transmission Area

If node does not receive HELLO message within the fixed time interval at each sequence purge it from NT. By this way update nearby routing node freshly. Mobile devices are deployed in the network region randomly and moves anywhere in the network using the random way point mobility model in non uniform manner. Source node initiates the data transmission by sending the first control packet to the destination.

In network layer, device looks for the routing table (RT) entry to forward the packet to the destination. If there is no route found in RT, then the device executes the route discovery process to identify the route to connect with the destination. When the source device wants to generate a new path to the final destination, the sender broadcast a route request packet to the entire network and it floods in many directions by its neighbors obtained from NT. When the neighbors collect the RREQ it produces a reverse path to the source device. By these way neighbor devices of next hop to the sender established. While forwarding the RREQ and RREP the reverse entry and forwarding entry is formed respectively. The middle devices re-broadcast the RREQ to their neighbors and update the device address as value[i] in their RT as

RTTable.count++
ΣRT += RTTable.value[i]

These message exchanges will be used to form the reverse path for route reply RREP from the destination. The devices collecting these packets are cached from source and when the link is disconnected by using this it sends RERR packet which holds information about devices that are unable to access. After constructing the path, the devices forward the data according to the constructed path from source to destination.

Due to mobility, if link failure occurs then the route reconstruction process is invoked to identify the alternate route to connect to the destination device. During data transmission, a packet either gets dropped or gets delayed due to network congestion.

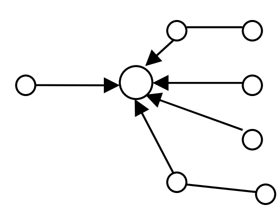

Figure 3. Router Congested Node

In transport layer, congestion control mechanism is operated to identify and to control the network congestion in base model using the dedicated parameters such as

Middle Packet Setback Distinction (MPSD) and temporary bits per second (TB). In this Initial MPSD = 0.
Update MPSD count++

The Sum of MPSD is computed using the formula

$$\Sigma MPSD \mathrel{+}= MPSDValue[i];$$

Average of MPSD is computed using the formula

$$AvgMPSD = \frac{\Sigma MPSD}{MPSDCount}$$

$$D\Sigma MPSDCount{+}{+}$$

$$D\Sigma MPSD \mathrel{+}= (AvgMPSD - MPSDValue[i])^2$$

Computed variance of MPSD

$$VarianceMPSD = D\Sigma MPSDCount / MPSDCount$$

For MPSD an upper bound (UNMPSD) and lower bound (LBMPSD) are defined as threshold points.

$$StddevMPS = \sqrt{VarianceMPSD}$$

$$LBMPSD = AvgMPSD - StddevMPSD$$

$$UBMPSD = AvgMPSD + StddevMPSD$$

If the current statistics of the network crossed these threshold boundaries then the network is marked as congested network.

if(MPSD < LBMPSD || MPSD >UBMPSD)

Enable congestion Flag = 1

Also the TB computed as

TBCount++

$$\Sigma TB \mathrel{+}= TBvalue[i];$$

$$AvgTB = \Sigma TB / TBCount;$$

$$D\,\Sigma TB \mathrel{+}= (AvgTB - TBValue[i])^2$$

MBS is computed based on idle packet state; success packet state and collision state at channel usage period and are also based on the number of transmissions as success, collision and idle states as given below. Channel usage (CU) estimation provides the present utilization of overall bandwidth consumption.

PI = pow((1-tow),n);

$$PS = N\neg(1 - \neg NI)^2$$

PC = 1 - Pidle - Psuccess;

TS = TData + TAck + SIFS + DIFS

TC = TData + Timeout + DIFS

TI = 1 - TSuc – Tcol

$$CU = \frac{PSxTS}{(PIxTI+PSxTS+PCxTC)}$$

$$MBS = \frac{(PSxTS + PCxTC)}{(PIxTI+PSxTS+PCxTC)}$$

Due to link failure and node failure, these parameters may provide false deviations as the congested state of the network. In order to avoid these false positive cases, the metrics are extended as follows, Completion Time (CT) is taken as

$D\Sigma CT \mathrel{+}= (AvgCT - CTValue[i])^2$,
Packet Loss Case (PLC) is computed as

$D\Sigma PLC \mathrel{+}= (AvgPLC - PLCValue[i])^2$, and Packet Delay (PD) along with the Medium Busy State(MBS).

$$PD = \frac{\Sigma PD}{PDCount}$$

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For these extended parameters, both lower and upper bound are defined as threshold limits which are used to classify the current network situation as congested or not. If it is congested then congestion flag is enabled.

Update MPSDadd(MPSD)

Update TBAdd(TB)

Enable CongestionFlag

Each parameter is computed using the statistical model related to the Gaussian distribution with mean and standard deviation. To compute these parameters, the corresponding information is maintained in the newly added header called TCP Congestion header (CH). CH is computed as the time difference between data packet sent and acknowledgment received.

AvgDelay = αAvgDelay + (1-α) x CHDS

CHMBS Time = Max(CHMBSTime,MBS)

CurrenMBSTime = CH MBSTime

CurrentDelay =CHDS

Source node attaches the send time information in CH, destination receives the information and copy it in CH of ACK message. Once the ACK message reaches the source end, it calculates the difference between the ACK packet received time and Data Packet sent time as CH. MPSD is computed as average delay. Transmission delay is computed as the time difference between the packets sent and packets received for both data and acknowledgment messages.

AvgDelay = α * AvgDelay + (1-α) * CHDS

CHMBS Time = Max(CHMBSTime,MBS)

CurrenMBSTime = CH MBSTime

CurrentDelay =CHDS

If (Enclosed Reaction == 1)

CHMBSTime = ƀ * AavgMBS + (1-b) *

CurrentMBSTime

CHDS = b *AvgDelay + (1-b) *CurrentDelay

CH NewRate = NewRate

PLC is computed using the packet sequence number difference at the destination end. It is calculated as the ratio between the number of packets not in order and the total number of packet transmitted. TB is computed as average number of bits transmitted in cycle duration.

Using the current data rate, data packet size and acknowledgment size, approximate channel occupancy time are computed for both data packet and acknowledgment packet. Back-off timeout period is also computed using the channel occupancy time, propagation delay and retransmission delay. Using the individual slot time for each node, idle probability, transmission successful probability and collision probability are estimated. The successful transmission duration is computed as the sum of Data duration, Acknowledgment duration, SIFS and DIFS periods.

The collision duration is computed as the sum of data duration, timeout period and DIFS periods. From the unit slot duration, idle slot period is computed using the difference between unit with sum of successful transmission duration and collision duration. MBS is computed using the ratio product of successful transmission duration and successful transmission probability with entire probability duration. MBS is computed as the ratio product of successful transmission duration with collision duration

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and successful transmission probability along with collision probability to the entire probability duration as delay.

If (PacketType == ACK)
If (CongFlag == 1)
Compute QueueDelay = CURRENT_TIME - CHTime
CH**Σ**Ds += QueueDelay
CHCountDs++
CHDS =CH**Σ**Ds /CHCountDs
If (AveDelay == 0)
AvgDelay =CHDS

The MBS is estimated in Mac layer and attached in the CH header field named MBS metric.

B. Fuzzy Inference System

The fuzzy inference system operates at destination node. After collecting delay, ongoing rate and MBS parameters, they are given as an input to the FIS based congestion detection system

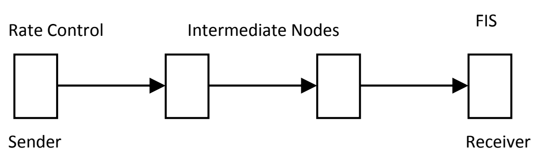

Fig.4 Fuzzy based Congestion Detection System

which employs fuzzification, rule set matching and detection of congestion state. In fuzzification stage, using the lower and upper bound the input parameters are converted into fuzzy linguistic variables.

FuzzyIinput = MBS Delay RS
FuzzyRule = RuleSetMatching (Fuzzy1,Fuzzy2,Fuzzy3)
CongState = CongestionState(FuzzyRule)
Set CongState

The linear relationship growth model is used in rule set formation since these parameters have uniform deviations in the parametric values. Identified fuzzy linguistics are compared with the rule set to compute the exact match in the rule set to select current state of congestion. Congestion states are classified as VLOW, LOW, MEDIUM, HIGH, VHIGH.

CongestionState (FuzzyRule)
if(rule == VLOW)(0,0.2)
if(rule == LOW) (0.2,0.4)
if(rule == MEDIUM) (0.4,0.6)
if(rule == HI)(0.6,0.8)
if(rule == VHI)(0.8,1.0)

If congestion is detected in the network state, then the CH is marked in the acknowledgment message. Once the acknowledgment packet with congestion notification reaches the source node, it reduces the current data rate to decrease the outgoing packet count.
If congestion state is VLOW or LOW then destination calculates new data rate using multiplicative increase. If congestion state is MEDIUM then destination calculates new data rate using additive increase. If congestion state is HI or VHI then it calculates new data rate using multiplicative decrease. It updates the new data rate and sends the acknowledgement packet to sender. Intermediate nodes increase or decrease congestion window size as per the new data rate they receive from the neighbor.

C. E-MODEL

E-model is the common ITU-T transmission rating model. This computational model can be used to measure the quality of Voice data that help ensure that users will be satisfied with end-to-end transmission performance. The

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main output of the model is a scalar rating of transmission quality.

It is the most commonly used method for predicting the quality of voice signal on user side.

The ITU-TG.107 defines the relationship between R and MOS as follows:

MOS = 1 for R<0

MOS = 1+0.035R+R(R-60)*(100-R) 7*0.000001 for 0<R<100

MOS = 4.5 for R>=100

The rating factor *R* is composed of the basic formula:
R = Ro – Is – Id – Ie-eff + A

Ro (signal to noise ratio) is a mathematical summary of how the voice levels compare to the different noise sources including circuit noise and room noise.
Id (delay impairments) is a mathematical summary of transmission delay, talker echo and sidetone.
Is (simultaneous impairments) considers non-optimum sidetone, quantizing distortion, overall loudness and other impairments which occur more or less simultaneously with the voice transmission.
Ie (equipment impairment) and A (Advantage Factor) are both single value quantities.
To assist with calculations, default values and permitted ranges have been established.

This calculates MOS value based on the given criteria[3].

The following table shows the relationship between R factor and user satisfaction.

R Factor	MOS	User Satisfaction
140	4.5	Very Satisfied
80	4.024	Satisfied
120	4.5	Very Satisfied
138	4.5	Very Satisfied
110	4.5	Very Satisfied

IV. RESULT ANALYSIS

Simulation compares the proposed approach CLHCC with FARCC and ADTCP by varying the number of nodes and simulation time. Results show that the performance of the proposed approach at network level and at user level. It outperforms well than the two approaches in terms of packet delivery ratio, end-end-delay, through put. It calculates user level quality called Quality of Experience (QoE) using E-Model. MOS values show that the proposed approach attains user dissatisfaction level in terms of quality.

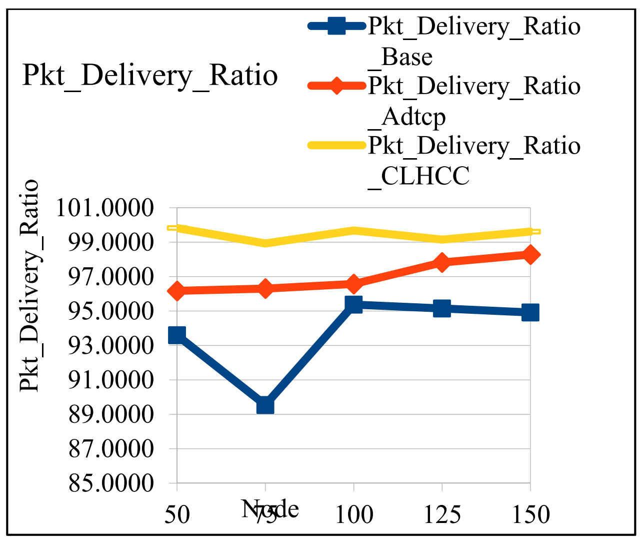

Fig 5 : Node Versus Packet Delivery Ratio

Figure 5 shows the comparison between three approaches based on the metric Packet Delivery Ratio. The graph clearly indicates that the proposed approach outperforms well than the existing approaches.

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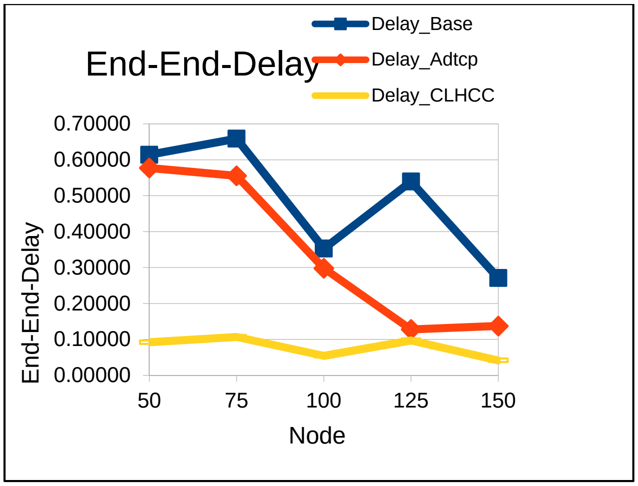

Fig 6 : Node Versus End-End-Delay

Figure 6 shows the performance of proposed approach in terms of End-End-Delay. It suffers from less End-End-Delay than the existing approaches.

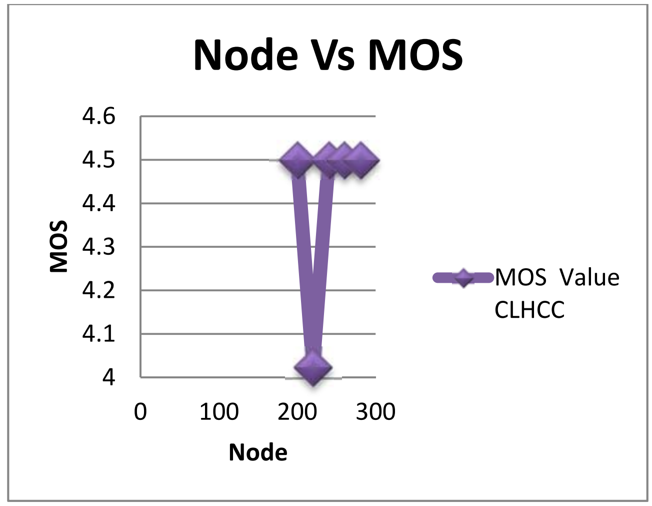

Fig: Node Versus MOS

Figure 7 shows that user level quality of Voice. Quality remains at same level as the performance increases.

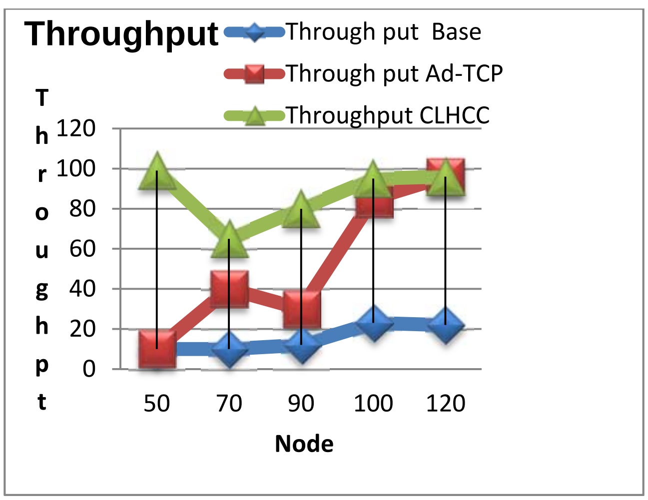

Fig 7 : Node Vs Throughput

Figure 7 shows the maximum through put attainment of proposed approach.

V. CONCLUSION

In Mobile Ad-hoc Network, MBS, TB, Delay distinction are used to measure the congestion level of the network. These investigations support us to change transmission rate. In this paper, we checked congestion level in MANET with Fuzzy Inference System to finalize the data rate and to control the flow of data at sender by the receiver dynamically. It also adjusts the congestion window as per the new data rate on the stamp field. Hence it supports both proactive and reactive approach to congestion detection and to control congestion. The proposed approach outperforms well for network level quality and user level quality than the existing congestion control approaches ADTCP and FARCC.

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